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EBU Guidelines for Multichannel Audio in DVB

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Contents

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1. Introduction

This document presents the EBU Guidelines for Multichannel Audio (MCA) transmissions in Digital Video Broadcasting (DVB).

These guidelines are intended for both broadcasters who provide multichannel audio transmissions over DVB-S, DVB-C or DVB-T as well as manufacturers who produce DVB Set-top boxes (STBs) and DVB Integrated Circuits (ICs).

The initiative for preparing this document has arisen from the Members' experience that many of the STBs currently available in European markets do not perform according to broadcasters' and consumers' expectations. They may not comply with the existing ETSI specifications such as TS 101 154 [1]. Consequently, our viewers and listeners may experience great difficulties in accessing the MCA services in a consistent and reliable manner.

The EBU agreed to act as a facilitator to help both broadcasters and manufacturers to remove the bugs and deficiencies related to MCA from the broadcast chain and the STBs in the market. Our objective is that millions of users Europe wide who are already able to experience MCA transmissions will be more satisfied with the quality of the broadcast services which are available today and will be available in the forthcoming years.

The present Guidelines document is based on the extensive evaluations of a number of commercially available DVB STBs. The EBU Focus Group B/MCAT (Multichannel Audio Transmission) has been mandated to test these STBs in order to assess their behaviour and performances in terms of MCA operational and technical quality[1].

This Guidelines document deals with the MCA services carried by both radio (audio-only) and television broadcasts. We used an extensive list of specific test bitstreams, which were designed to critically assess the technical performance of the STBs under test. It was of course not possible to test all STBs in the market but in the limited time available, we managed to examine the performance of a representative sample of commercially available STBs.

The document is based on the collective knowledge and experience of various European broadcasters, conducting trials, pilots and permanent services with 5.1-multichannel audio transmission over DVB.

The summary of the STB evaluations is given in **Appendix 1** to this document. **Appendix 2** gives a descriptive list of the test bitstreams used.

In order to facilitate our discussions with the STB manufacturers, the EBU has already produced a document Tech 3307 [2] that outlines broadcasters' service requirements for future free-to-air

[1] B/MCAT has produced a document entitled “Present Multichannel Audio, Radio and TV Broadcast Practices and Operations in Europe”. The document is available to EBU members only.

HDTV receivers. Whilst our collaboration with manufacturers has already proved to be useful and necessary, further cooperation is required between broadcasters and manufacturers in order to remove the remaining problems and misunderstandings.

Feedback on this document is welcomed from any of the parties concerned.

2. Scope and Basic Requirements.

The basic requirements for implementation of MCA in DVB systems are already given in ETSI TS 101 154 [1]. This document provides sufficient information about encoding of MPEG-1 Layer 2 stereo audio in DVB broadcast bit-streams, and for decoding this bit-stream in the STB. In addition, it also provides information about the implementation of multichannel audio bit-streams in DVB compliant transport streams.

The present Guidelines document complements document [1]. It provides some additional requirements and recommendations concerning the implementation of particular audio features required to enhance the user experience. It is strongly advised that both broadcasters and STB manufacturers take into account these Guidelines, in order to ensure a consistent behaviour of the STBs and satisfactory reception of the broadcast services to the end user.

The present Guidelines mainly refer to the functionalities and behaviour of the multichannel audio (MCA) systems. Where necessary, we also address the system aspects of current standard television (SDTV) and radio-only DVB services. Our findings also apply to future HDTV services, unless stated otherwise. However, it might be necessary to consider specific requirements for HDTV audio services in the future.

As required by [1], a STB shall be able to decode a stereo signal as a minimum. For this stereo service, at least the MPEG-1 Layer 2 coding format shall be implemented in the STB. This stereo signal shall be available at the analogue out at all times.

On S/PDIF interface, either the MPEG-1 Layer 2 decoded PCM signal or the MCA bitstream shall be available and transparently routed to the digital interface at all times. If more than one MCA bitstream is transmitted, the appropriate stream should be user-selectable.

For multichannel services, a coding system according to [1] shall be used.

MCA in television and MCA in radio have similar operational functionalities and should thus be handled in the same way.

3. Audio Navigation

According to [1], there may be more than one audio bitstream belonging to a given service. For example, an MCA signal can be simulcast with the MPEG-1 Layer 2 Stereo audio.

If more than one audio bitstream is present, it is up to the user of the STB to select the appropriate one. This functionality should be readily accessible. The user should be able to select either Stereo or MCA as a "first choice" preference. Should the user select MCA as their first choice, then this will be automatically selected by the STB whenever a MCA service is present. For radio services it would be helpful if the user could choose between the different service components without the need of OSD (on-screen display) navigation. For example, he/she could use a toggle button together with indication of the selected audio format on the front panel of the STB.

Where MCA is one of several components in the service bitstream, the relative order in which these various components are referenced in relevant PSI and SI tables receivers may change from one stream to another as the result of operational configuration changes at the head-end. Receivers shall accommodate this.

Audio bitstreams may be configured in different audio formats, such as 2.0, 5.0, 5.1 and others. This audio bitstream is passed unchanged onto a "digital bitstream out" interface of the STBs.

Decoding of these bitstreams takes place in a separate decoding device (usually an "A/V receiver"), which is connected to the STB.

If language information is available, the STB shall use this information to select the user's preference.

Requirements about the selection of the audio signal for physical interfaces are described in section 11, "*Physical interfaces of the set top box for audio*".

4. Audio/Video Synchronization

Synchronization of audio and video components is one of the most frequent problems and one of the most annoying. It is often called "lip-sync". The DVB system uses a time stamp mechanism that maintains synchronism between audio and video of the transmitted signal. Timing between encoder and decoder should be maintained within 1ms.

The STB should attempt to preserve audio/video synchronization of the received signal. To this end, the STB needs to implement appropriate buffer management and sufficient robustness of the A/V synchronization mechanism.

4.1 Buffer management

Buffer management in the STB shall be in compliance with the requirements of the respective audio formats used in DVB [1].

Note: Various audio coding formats may require different audio buffer sizes in order to achieve audio/video synchronization. For example, document ITU-R BS.1196-1 entitled "Audio coding for digital terrestrial television broadcasting" references different buffer sizes for MPEG-1 Layer 2, Dolby AC-3 for ATSC, Dolby AC-3 for DVB.

A correct adjustment of the buffer operating point is another pre-condition for maintaining robust and reproducible A/V-sync in typical operations.

4.2 Robustness of A/V synchronization.

The implementation of A/V synchronization in a STB must be robust, i.e. it does not change over time or due to the different stimuli which may occur in practice. In general, STBs may have different strategies to ensure A/V sync. There are STBs which apply a variety of different strategies dependant on the context.

Correct A/V sync shall be maintained under the following conditions:

Cold start.

- selection of a service after new channel acquisition (inc. receiver installation).
- selection of the last-used service on switch-on.

Channel change.

- channel change within same multiplex.
- channel change from an alternative multiplex.
  - with no change of modulation.
  - with a change in modulation, e.g. 64QAM to16QAM in DVB-T or 4-PSK to 8-PSK in DVB-S(2).
- from an alternative delivery platform (e.g. DTT to DSAT).

Alternative navigation.

- selection of the service from a data application (e.g. quarter-screen inset in an application to full-screen).
- navigation from and back to a parent service

Audio service selection

- user-selected change of audio service, for example change from stereo to MCA, or change of language.
- automatic change of audio service, for example due to the addition or removal of currently selected MCA stream.

Changes of service.

- time-exclusive use of bit-rate (e.g. service X closes at 19.00 to be replaced by service Y on the same PIDs).
- once-off changes of component (e.g. “we are pleased to announce that from 01/03/2007 at 19.00 hours onwards MCA will be available on service Y”).
- scheduled changes of component.
  - signalled in PSI (i.e. the PMT) only.
  - signalled in SI (e.g. $EIT_{p/f}$, EIT_{sch}) only
  - signalled in PSI & SI.

Error recovery.

- from uncorrected transmission errors.
- from a redundancy switch at the head-end of the whole TS.
- from a redundancy switch at the head-end of coders supporting the same service.
- from a discontinuity of the programme clock reference (PCR) for that service

Other factors.

Whilst changes of service component parameters may not necessarily have a direct impact on service synchronization, receiver manufacturers need to be able to manage changes to coding parameters such as bit-rate, coding mode etc. on an event basis.

Successful synchronization strategies imply that the programme clock reference (which the timestamps for any time-critical service component refer to) may be carried in any of the options below:

- a video component (if any) of the selected service,.
- a video component of another service in the same multiplex,.
- one of the audio components of the selected service,.
- an audio component of another service in the same multiplex or.
- a separate PID (e.g. common PCR to several services in the same multiplex).

5. Compensation for A/V delay in the receivers

5.1 Baseband analogue and digital Audio output

Audio shall be synchronous with video (for the same television programme) at all control junctions of the transmission chain; audio/video synchronization in the set-top box (STB) must be within a tolerance of -5ms (audio early) to +15ms (audio late) between decoded video and decoded audio outputs which shall not drift after synchronization .

5.2 IEC61937 output

Where audio leaves the STB in an encoded form (such as in IEC61937 [5] outputs), the STB should compensate for the decoding latency of a reference decoder for the respective audio coding system, such that the decoded output of a reference decoder would be -5ms to +15ms with respect to the decoded video.

5.3 Delay due to flat panel displays and projectors.

Additional delay in the video path introduced by the display device shall be compensated by suitable mechanisms (i.e. internal audio portion of the display itself, external audio processor or internal pre-compensation in the set-top box). This shall apply to all selected audio components available in the service.

6. Audio services and signalling

Signalling for audio services within DVB are described in [1] and [2]. However, these documents allow a very large range of possibilities. Listed below are several of the specific requirements for multichannel audio services that are not described explicitly in these two documents.

6.1 Service types

Some broadcasters may transmit audio only services, signalled in SI [3] as radio, which contain both MPEG-1 Layer 2 and MCA.

Some broadcasters may however transmit audio only services, signalled in SI as radio, which contain only MCA.

6.2 Number of components per service

Some broadcasters may not transmit a permanent, multichannel audio service on a particular channel. These channels will add the MCA service at some time before the start of a particular programme and will terminate this stream some time after the end of the programme. Receivers should detect the presence of the MCA component and apply adequate means to inform the user or apply actions. (e.g., if pre-selected by the user, the STB should switch to this component automatically). For transmission, changes in the availability of an MCA component are indicated by changing PMTs[1] (including their version number) as defined by MPEG.

6.3 Content of a single component

Some broadcasters may choose to change the data rate of audio services within their broadcasts according to the number of audio channels encoded within the (MCA) component and/or the

[1] Programme Map Tables

sampling rate. Continuous audio presentation with a minimum of audible artefacts shall be achieved by the decoding system.

6.4 Changing of video and audio format

Some broadcasters may dynamically change picture format (e.g. 4:3 to 16:9) and audio format (e.g. stereo to MCA) at the same time. Whilst some STBs perform this change correctly, others experience discontinuity of video and audio. Such service interruption should be avoided.

In a new generation of boxes it should be possible to receive and decode additional audio services for the visually or hearing impaired (or alternative languages or commentaries) at an efficient bit rate and within current broadcast infrastructures. The ability to mix these additional services within the receiver would be beneficial.

Note: All information on coding modes and bitrates is carried in the audio elementary stream and the set top box should make adequate use of it and may signal this information to the user. Additional information may optionally be available from the EIT.

7. Audio ruggedness

STBs that implement the specified buffer management model and MPEG time stamps should not suffer from audio drop-outs. On any occasion where drop-outs do occur, for example due to a reception problem, the receiver should recover gracefully with minimum audible disruption and maintaining A/V sync.

8. Resilience in the case of transmission errors.

Receivers should handle transmission errors gracefully. This means that when a corruption or dropout at the transport packet or PES level occurs, the receiver shall recover the corrupted service within 1 second. Corruption within only one part of the programme (e.g. video, audio, and multichannel audio) should only affect that part – i.e. a dropout in video should not affect the audio.

9. Display radio text and/or SI derived programme-related information.

It was experienced that some STBs muted their audio outputs when a radio text service was signalled as being present. It was discovered that this anomaly is due to incorrect implementation of a specific integrated circuit.

STBs are required to implement the radio text feature appropriately such that audio is not muted when radio text is being displayed.

10. PVR functionalities in the context with multichannel audio.

Personal Video Recorders (PVRs) are required to support the following functions:

- Signalling of MCA at the time of PVR programming shall not be required for the MCA to be recorded. For example, some services may use dynamic changes in PMT to signal the presence of MCA. Therefore, the MCA stream may not be present at the time of programming.
- All audio and video components of the service present during the recoding shall be recorded. Therefore, the PMT should be examined continuously to identify and record any additional components.

- It is recommended that the PVR recording shall follow the program event ID for precise scheduling of the recording. Broadcasters should provide up-to-date EIT for this purpose.
- Playback of audio with video shall maintain A/V sync.

11. Physical interfaces of the set-top-box for audio

The following audio interfaces between receiver and home audio equipment shall be provided:

- Stereo audio shall be available on a pair of analogue stereo outputs.
- Suitable digital output for bitstream out shall be available on IEC 61937 and/or HDMI and/or IEEE 1394 (known variously as FireWire, iLink etc.).

Note: Where compatibility with legacy receivers is a commercial requirement, IEC 61937 shall be mandatory.

- The STB should be able to provide the user with the following choices: stereo PCM signal, multichannel PCM signal (if available), or encoded bitstream on its digital output.

Interfaces following the IEC61937 (coaxial and optical S/PDIF) shall set Pa, Pb, Pc and Pd words correctly. The Pa word shall be aligned in the left sample of the output interface. The spacing between Pa words shall be set correctly, for example 1536 samples for Dolby Digital at 48kHz. The receiver shall send pause data-bursts to the home audio equipment in cases where there is no audio data available due to transmission interruptions. This maintains synchronization of the decoder in the home audio equipment and allows quick reacquisition of the audio signal.

The IEC 61937 interface shall be compatible with the audio formats defined in [1].

It would be helpful to have always a stereo signal at the analogue output coming either from the MPEG stereo or from a MCA downmix.

Special attention should be given to the output jitter of the digital audio interface, which is used for the bitstream out. The maximum jitter should be limited to the relevant boundaries, such as rise and fall times, intrinsic jitter and jitter gain and peaking, specified in [4].

12. Summary and Conclusions.

This document summarizes the experience of those EBU members who have already started regular MCA operations, or are in the process of doing so. We have tested a number of commercially available DVB STBs and have detected several problems, which are reported in the present Guidelines document.

Most severe areas of concern encountered are given in **Appendix 1** of this document. These areas relate mainly to the ability to tune, select and identify multichannel audio in radio and television operations. Most problems happen when some changes occur in the multiplex. These may involve data rate changes, changes in numbers and type of audio channels, etc. Particular attention was paid to the A/V synchronization, whereby several anomalies were observed in the practical implementation of STBs.

The EBU members are strongly committed to comply with the existing DVB standards that specify the tools for MCA. The EBU Guidance will help those EBU members who have not started to implement MCA in a compliant way.

Our experience is that manufacturers have not paid sufficient attention to the quality of MCA in the implementation of the STBs in the market and consequently many anomalies have been detected. The manufacturers should take into account these Guidelines and implement boxes in such a way as to minimize possible unexpected behaviour of STBs concerning the rendering of MCA.

In order to accommodate any coding scheme standardized in TS 101 154, STBs should be able to pass through any bitstream that appears on the digital output. Using this transparency approach allows broadcasters to use the different current and future coding formats.

As this document is of common interest to both broadcasters and manufacturers, it will be disseminated widely to both communities. We plan to forward it to all EBU members, DVB Forum, EICTA and other bodies. It is important that comments and observations are obtained from as many different organizations as possible and that a fruitful dialogue continues with all of them.

References:

[1] ETSI TS 101 154 V1.7.7 (TM 1214 rev. 27): Implementation guidelines for the use of Video and Audio Coding in Broadcasting Applications based on the MPEG-2 Transport Stream.

[2] EBU - Tech 3307: Service Requirements For Free-To-Air High Definition Television Receivers, EBU; November 2005

[3] ETS 300 468 V1.7.1 (2005-08) - Specification for Service Information (SI) in DVB systems

[4] IEC 60958-1, “Digital audio interface – Part 1: General”, International Electrotechnical Commission, IEC, Second edition, 2004-03

Acronyms used

EIT Event Information Table

PID Packet Identifier

PMT Programme Map Table

PSI Programme Specific Information

SI Service Information

Appendix 1: Test results on STB MCA performance

A1.1 Introduction

During the trials and launches of multichannel radio and television services EBU B/MCAT members became aware of a number of performance problems with set top boxes.

A set of test bitstreams were created, in discussion with B/MCAT members, to examine these problems in a quantitative manner to understand better what is possible now, and what would be possible in the future, for multichannel audio transmissions. To date these tests have been predominantly conducted on Dolby Digital services but experience with the broadcast trials has included DTS and MPEG services as well. The range of test bitstreams is illustrated in the list given at the end of this document (see **Appendix 2**).

The bitstream tests have been conducted on a representative range of commercially available STBs (in total 25 STBs) produced by several manufacturers used by EBU members for their multichannel audio trials. In some cases obvious problems were corrected in collaboration with the manufacturers prior to the tests commencing.

These tests were conducted by Dolby, IRT, WDR and SVT under the auspices of the EBU B/MCAT group.

As an outcome of these tests, Dolby has launched a comprehensive range of bitstream tests for their licensees and it is working with manufacturers to refine the necessary test schedule.

The purpose of this document is to help broadcasters and manufacturers in facilitating the introduction of multichannel services.

A1.2 Test procedure

The test equipment comprised an ASI interface card Dektec DTA-140, the QPSK modulator Dektec DTA-107 for DVB-S boxes and a COFDM modulator Broadcast Technology DTMD 1000 for DVB-T boxes. The software used was.

- DTC-300 StreamXpress (transport stream player)
- DTC-320 StreamXpert (transport stream analysis and recording)
- Dtloop (utility for connection between the ASI card and the modulator)

In the tests, the stream player repeatedly replayed each of the bitstreams in a loop under user control. The tests were conducted in controlled environments at each location. Included in the test sequence were a limited number of STBs that were tested at more than one location – in each case, similar results were obtained from the different locations.

A1.3 Areas of concern

The basis of the tests was to explore the following areas of concern:

1. Ability to tune, select and correctly identify multichannel audio radio and television services irrespective of the ordering of PID values and the presence of video and/or stereo MPEG-1 L2 audio.
2. What happens to the stereo analogue audio output when multichannel audio is selected.
3. Ability to switch multichannel audio streams on or off within a service for individual programmes or groups of programmes.

4. Ability to use a variety of data rates for the multichannel audio service and to dynamically change these data rates as the number of audio channels in the multichannel audio service changes.
5. Ability for the STB to respond to errors within the transmission at both transport packet and elementary stream levels.
6. A/V synchronization of both stereo and multichannel audio on analogue and digital outputs.
7. Formatting of the S/PDIF and/or optical digital output.
8. Interaction of radio text with stereo and multichannel audio.
9. Ability to accommodate different sample rates (48 kHz, 44.1 kHz and 32 kHz) within a transmission.

A1.4 Test results

Identification of Television and Radio services

Each STB correctly identified TV services that contained video, MPEG-1 L2 audio and DD audio, irrespective of the ordering of the PIDs.

In the absence of the MPEG-1 L2 signal, a minority of boxes either failed to tune to this TV service or failed to output the DD audio.

For those bitstreams providing audio only (Radio) service the following results were obtained.

- One STB failed to tune to any radio only service, including services with dummy video PIDs
- One STB failed to tune to services that did not contain MPEG-1 L2 audio
- Three STBs tuned to the services but both the stereo and multichannel outputs suffered extensive drop-outs.
  - For one STB such drop-outs could be corrected by reselecting the audio service.
- One STB detected the presence of DD audio but did not output the multichannel audio

Stereo analogue audio output

For each STB the stereo output was silent once the DD audio was selected. (NB outside the range of boxes subjected to these specific tests, it is known that there are other STBs which maintain the stereo analogue audio output even when the DD audio is selected.).

Comment: The expectation of broadcasters is that the stereo analogue audio output should be maintained whether or not DD audio has been selected. It is further expected that if DD is broadcast without a stereo elementary stream, then the STB should provide a downmixed and decoded stereo signal on the analogue output.

Switching audio streams on and off within a TV service

In the case of switched streams with corresponding SI changes, when the DD disappeared each of the STBs correctly switched to the MPEG audio. However, very few STBs automatically switched back to the DD stream when it reappeared.

Comment: This automatic reacquisition of DD is important for users in the context of PVRs. Even in the absence of a PVR, without this feature, users would have to manually reselect DD after each discontinuity.

It was further noted that the disappearance and reappearance of DD caused an audible glitch on both the MPEG audio, when selected, and on the analogue output.

When MPEG audio had been selected by the user and the service disappeared, most STBs went to silence but one STB switched to DD. In this particular case the STB stayed switched to DD when the MPEG service reappeared, when it should have switched back to the user-selected MPEG.

In the case where no corresponding SI changes were made (i.e. DD was always signalled as an available service) the automatic behaviour of the STBs was the same as listed above. However, each of the STBs allowed the user to select DD even if it was not available, causing the user to hear silence.

Comment: The behaviour of the STBs in allowing users to select DD when not there but when signalled in the SI is correct: the fault is at the point of transmission in allowing an illegal bitstream/SI combination.

Dynamic changes in DD audio data rate

Two types of test were conducted; firstly sequencing through the full range of DD data rate and secondly switching between just the two most commonly used data rates of 192 and 384 kbit/s (stereo to 5.1).

The responses of the STBs were broadly correct.

Some STBs would not decode DD at a higher bitrate than 512 kbit/s.

Comment: Bitrates up to 640 kbit/s are legal and could be used by broadcasters.

Some STBs gave a short mute of the DD output at the point of the data rate change. Comment: As such changes are likely to take place at programme junctions, this behaviour is deemed acceptable under these circumstances, but should be minimized.

One STB would not output audio for bitrates between 32 and 56 kbit/s.

Comment: These data rates would not normally be used for DD.

Robustness to errors

The tests included damage to transport packages alone and elementary streams alone. In each case the damage was applied to video or MPEG audio or DD audio in sequence. In such cases, it is natural to expect some transient disruption on the output from any STB.

In some STBs there was no discernable disruption to audio or video services.

At no time did an error on one of the audio services cause problems with either the other audio service or the video service.

For one STB disruption to the video caused problems with both the MPEG audio and the DD audio.

One STB took approximately 5 seconds to recover from an error to the DD elementary stream. For most of the STBs the recovery time was found to be less than 0.5 s.

A/V synchronization

The relative timings of audio and video outputs was found to be very variable, both between different STBs and for each STB under the different conditions in which it was tested, e.g. recovery from changes of channel, recovery from errors and general acquisition of streams.

In what follows, all timing measurements were made on the S/PDIF output. In all cases where precise timing figures were measured for A/V offset, the offset exceeded the recommended tolerances, -5ms (audio early) to +15ms (audio late).

For most STBs, there was inconsistency in the measured A/V timing even when the same service was being acquired several times over.

In most cases, if a STB has audio advanced on video for MPEG, it also has DD advanced on video. (Similarly for audio delayed on video.)

Within any particular STB the variation in A/V synch about its average value is limited to +/- 20ms.

Where two different bitrates were tested there was generally no significant difference in the A/V synch for a particular STB.

	Range of measured A/V offsets	
	MPEG-1 L2	DD[1]
Earliest/latest offset	-51 to +70 ms	-310 to +40 ms
Typical figures (80% of STBs)	-10 to +30 ms	-20 to +30 ms

S/PDIF output

Most STBs gave consistent signals on the S/PDIF output. However, one STB took several seconds for the output to stabilize (irregular data burst timing) after acquisition of a service.

Radio text

On radio (audio-only) services, some STBs using a specific integrated circuit muted their audio outputs when a radio text service was signalled as being present but the last bytes were set to zero (compliant with the standard). After further investigation, it was determined this was due to an incorrect implementation in this IC.

Alternative sample rates

At 48 kHz no problems were encountered, except in the circumstances given above.

At 44.1 kHz some boxes gave dropouts on MPEG and/or DD audio, whilst some STBs failed to give a DD output.

At 32 kHz these problem got worse and more STBs were affected. In addition, one STB changed audio pitch on the MPEG audio.

Conclusions

Due to the pressure of time and the wide range of STBs on the market, these results are just a snapshot of the performances of a small range (25) of representative STBs at the time of the tests. The performance of DD multichannel STBs, only, was tested. The equivalent parameters for other multichannel audio systems have not yet been evaluated. Any manufacturer who wishes to see the full results for their own STB will be supplied them on request to Dolby.

For Radio broadcasters, the biggest issue is in the inability of some STBs to acquire and identify a Radio service.

For Television broadcasters the three most significant problems are the variability and range of A/V synch, the inability of the STBs to revert to the user-selected audio format automatically, and the absence of audio on the analogue outputs when a multichannel audio is selected.

[1] With DD one would expect an earlier offset to account for an external decoder. The measured value does not include any adjustment for this offset.

Appendix 2: Descriptive list of the test bitstreams.

A2.1 Identification of Television and Radio services

TV service streams with different PIDs

TV service streams with MPEG-1 L2 and Dolby Digital audio and streams with Dolby Digital only

Radio service streams with MPEG-1 L2 audio, Dolby Digital audio and both MPEG-1 L2 and Dolby Digital audio

Radio service streams with dummy video PID (i.e. signaled in SI but no video elementary stream)

Switching audio streams on and off within a service

TV services with either Dolby Digital audio added and removed or MPEG-1 L2 audio added and removed. All SI tables updated.

TV services with either Dolby Digital audio added and removed or MPEG-1 L2 audio added and removed. All SI tables are left unchanged.

Dynamic changes in audio data rate

TV and radio services covering the complete range of Dolby Digital data rates.

TV and radio services switching two common data rates such as 384 kbit/s and 192 kbit/s or 448 kbit/s and 256 kbit/s.

Robustness to error

Transport packet and elementary stream corruptions and drop-outs

A/V synchronization

TV services at two different data rates for A/V sync measurement

Alternative sample rates

TV services with MPEG-1 L2 and Dolby Digital audio at 48 kHz, 44.1 kHz and 32 kHz sample rates.

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